20

CLAIMS

1. A method of filtering a speech signal, **characterized** by the steps of

providing a filter (404) suited for reduction of distortion caused by speech coding;

estimating acoustic noise in said speech signal;

adapting said filter in response to the estimated acoustic noise to obtain an adapted filter; and

applying said adapted filter to said speech signal so as to reduce acoustic noise and distortion caused by speech coding in said speech signal.

2. A method as defined in claim 1, wherein said step of adapting said filter involves adjusting filter coefficients of said filter (404).

3. A method as defined in claim 2, wherein said steps of estimating, adapting and applying are performed for portions of said speech signal which contain speech as well as for portions which do not contain speech.

4. A method as defined in any of claims 2 or 3, wherein said filter (404) includes a short-term filter function designed for attenuation between spectrum formant peaks of said speech signal and wherein said filter coefficients include at least one coefficient that controls the frequency response of said short-term filter function.

5. A method as defined in claim 4, wherein said filter (404) includes a spectrum tilt compensation function and wherein said filter coefficients include at least one coefficient that controls said spectrum tilt compensation function.

6. A method as defined in any preceding claim, wherein acoustic noise in said speech signal is estimated as relative noise energy (SNR) and noise spectrum tilt.

7. A method as defined in any of claims 2-6, wherein said step of adapting is performed by selecting values for said filter coefficients from a lookup table (430), which maps a plurality of values (432) of estimated acoustic noise to a plurality of filter coefficient values (434).

8. A method as defined in any preceding claim, wherein said steps of estimating, adapting and applying are performed after a step of decoding said speech signal.

9. A method as defined in any one of claims 1-7, wherein said steps of estimating, adapting and applying are performed before a step of encoding said speech signal.

10. A method as defined in any preceding claim, wherein said speech signal comprises speech frames and wherein said steps of estimating, adapting and applying are performed on a frame-by-frame basis.

11. A method as defined in claim 7, further comprising the initial steps of generating said lookup table by:
    adding different artificial noise power spectra having given parameter(s) of acoustic noise to different clean speech power spectra;
    optimizing a predetermined distortion measure by applying said filter (404) to different combinations of clean speech power spectra and artificial noise power spectra; and

for said different combinations, saving in said
lookup table those filter coefficient values, for which
said predetermined distortion measure is optimal,
together with corresponding value(s) of said given
5   parameter(s) of acoustic noise.

    12. A method as defined in claim 11, wherein said
predetermined distortion measure includes Spectral
Distortion (SD).
10

    13. A method as defined in claim 11 or 12, wherein
said given parameters of acoustic noise include relative
noise energy (SNR) and noise spectrum tilt.

15   14. A method as defined in claim 10 when dependent
on claim 6, comprising the further steps, after said step
of estimating acoustic noise, of
    deciding whether the estimated relative noise energy
for a current speech frame is below a predetermined
20 threshold; and
    if so, not performing said steps of adapting filter
coefficients and applying said filter, and instead per-
forming energy attenuation on the current speech frame so
as to suppress acoustic noise in a speech pause.
25

    15. A speech filtering device (400) for a speech
signal, **characterized** by
    a filter (404) suited for reduction of distortion
caused by speech coding;
30   means (410) for estimating acoustic noise in said
speech signal; and
    means (420, 430) for adapting said filter in
response to the estimated acoustic noise,
    wherein said filter, when applied to said speech
35 signal, reduces acoustic noise and distortion caused by
speech coding in said speech signal.

16. A speech filtering device as in claim 15, wherein said means (420, 430) for adapting said filter (404) is arranged to adjust filter coefficients of said filter in response to the estimated acoustic noise.

17. A speech filtering device as in claim 16, wherein said means for estimating, said means for adapting and said filter are arranged to operate on portions of said speech signal which contain speech as well as on portions which do not contain speech.

18. A speech filtering device as in claim 16 or 17, wherein said filter (404) includes a short-term filter function designed for attenuation between spectrum formant peaks of said speech signal and wherein said filter coefficients include at least one coefficient that controls the frequency response of said short-term filter function.

19. A speech filtering device as in any of claims 15-18, wherein said means (410) for estimating acoustic noise is arranged to estimate it as relative noise energy (SNR) and noise spectrum tilt.

20. A speech filtering device as in any one of claims 16-19, wherein said means (420, 430) for adapting said filter (404) comprises a lookup table (430), which maps a plurality of values (432) of estimated acoustic noise to a plurality of filter coefficient values (434).

21. A speech filtering device as in any one of claims 15-20, wherein said speech signal comprises speech frames and wherein said means for estimating, said means for adapting and said filter are arranged to operate on said speech signal on a frame-by-frame basis.

22. A speech decoder comprising a speech filtering device according to any one of claims 15-21.

23. A speech decoder as in claim 22, wherein said
5   speech decoder is a CELP decoder.

24. A speech codec comprising a speech decoder according to any one of claims 22-23.

10   25. A speech transcoder comprising a speech decoder according to any one of claims 22-23.

26. A computer program product directly loadable into a memory (242) of a processor (240), where the
15   computer program product comprises program code for performing the method according to any of claims 1-14 when executed by said processor.

27. An integrated circuit, which is adapted to
20   perform the method according to any of claims 1-14.

28. A module, which is adapted to perform the method according to any of claims 1-14.

25   29. A station (200) for a mobile telecommunications network (110), comprising at least one of a speech filtering device according to any one of claims 15-21, a speech decoder according to claim 22 or 23, a speech codec according to claim 24, a speech transcoder
30   according to claim 25, an integrated circuit according to claim 27 or a module according to claim 28.

30. A station as in claim 29, wherein the station is a base station (104, 104').
35

31. A station as in claim 29, wherein the station is a mobile terminal (100, 100').